

CLAIMS

[1] A sampling rate converter comprising:

an up sampler for inserting $U-1$ zero points between sample signals and raising a sampling frequency U -fold,

5 a convolution processing unit including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler, and

a linear interpolation block for selecting two
10 points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation, wherein

the FIR filter of the convolution processing unit
15 is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes a filter coefficient, and a transmission function $H(z)$ is associated with a transmission function $Z(z)$ of a pre-filter, and

20 the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[2] A sampling rate converter as set forth in claim 1,
25 wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by

performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[3] A sampling rate converter as set forth in claim 1,
5 wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm considering a frequency response of the pre-filter.

[4] A sampling rate converter as set forth in claim 1,
10 further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[5] A sampling rate converter as set forth in claim 1,
15 further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

20 [6] A sampling rate converter comprising:

an up sampler for inserting $U-1$ zero points between sample signals and raising a sampling frequency U -fold,

a convolution processing unit including an FIR filter and performing predetermined convolution
25 processing with respect to an output signal of the up sampler, and

a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear

5 interpolation, wherein

the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes a filter coefficient, and

10 the filter coefficient is set by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[7] A sampling rate converter as set forth in claim 6,
15 wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point.

[8] A sampling rate converter as set forth in claim 6,
further including a low bandpass filter preventing an
20 aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[9] A sampling rate converter as set forth in claim 6,
further including a low bandpass filter preventing an
25 imaging component from occurring and a non-original frequency component from occurring when the sampling

frequency of the input is higher than a sampling frequency of the output.

[10] A sampling rate converter comprising:

an up sampler for inserting $U-1$ zero points between
5 sample signals and raising a sampling frequency U -fold,

a convolution processing unit including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler, and

10 a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation, wherein

15 the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes a filter coefficient, and a transmission function $H(z)$ is associated with a transmission function $Z(z)$ of a pre-
20 filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

25 [11] A sampling rate converter as set forth in claim 10, wherein the filter coefficient is set based on an

amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

5 [12] A sampling rate converter as set forth in claim 10, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point and considering a frequency response of the pre-filter.

10 [13] A sampling rate converter as set forth in claim 10, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

15 [14] A sampling rate converter as set forth in claim 10, further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling
20 frequency of the output.

[15] A sampling rate converter comprising:

a plurality of convolution processing units including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing the
25 convolution processing of input sample signals and the poly-phase filters decomposed to the poly-phases,

a plurality of up samplers for inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold,

5 an adding means for generating a signal after adding all signals by adjusting a propagation time of output signals of the plurality of up samplers, and

 a linear interpolation block for selecting two points of samples with respect to the signal by the
10 adding means and finding the value at the required position from the linear interpolation, wherein

 the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a
15 transmission function $H(z)$ is associated with a transmission function $Z(z)$ of a pre-filter, and

 the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the
20 pre-filter.

[16] A sampling rate converter as set forth in claim 15, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a
25 desired characteristic in relation to a frequency response of the pre-filter.

[17] A sampling rate converter as set forth in claim 15,
wherein the weighted approximation is performed with
respect to a desired characteristic using a Remex
Exchange algorithm considering a frequency response of
5 the pre-filter.

[18] A sampling rate converter as set forth in claim 15,
further including a low bandpass filter preventing an
aliasing component from occurring and folding from
occurring when the sampling frequency of the input is
10 lower than a sampling frequency of the output.

[19] A sampling rate converter as set forth in claim 15,
further including a low bandpass filter preventing an
imaging component from occurring and a non-original
frequency component from occurring when the sampling
15 frequency of the input is higher than a sampling
frequency of the output.

[20] A sampling rate converter comprising:

a plurality of convolution processing units
including pre-phase filters obtained by poly-phase
20 decomposing a predetermined FIR filter and performing
convolution processing of input sample signals and poly-
phase filters decomposed to poly-phases,

a plurality of up samplers for inserting U-1 zero
points between output signals of corresponding the
25 convolution processing units and raising the sampling
frequency U-fold,

an adding means for generating a signal after adding all signals by adjusting a propagation time of output signals of the plurality of up samplers, and

a linear interpolation block for selecting two
5 points of samples with respect to the signal by the adding means and finding the value at the required position from linear interpolation, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, and an
10 impulse response becomes the filter coefficient, and

the filter coefficient is set by performing the weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

15 [21] A sampling rate converter as set forth in claim 20, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point.

[22] A sampling rate converter as set forth in claim 20,
20 further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[23] A sampling rate converter as set forth in claim 20,
25 further including a low bandpass filter preventing an imaging component from occurring and a non-original

frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

[24] A sampling rate converter comprising:

- 5 a plurality of convolution processing units including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases,
- 10 a plurality of up samplers for inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold,
- an adding means for generating a signal after
- 15 adding all signals by adjusting a propagation time of output signals of the plurality of up samplers, and
- a linear interpolation block for selecting two points of samples with respect to the signal by the adding means and finding the value at the required
- 20 position from linear interpolation, wherein
- the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, an impulse response becomes the filter coefficient, and a transmission function $H(z)$ is associated with a
- 25 transmission function $Z(z)$ of a pre-filter, and
- the filter coefficient is set by performing

weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

[25] A sampling rate converter as set forth in claim 24,
5 wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

10 [26] A sampling rate converter as set forth in claim 24, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point and considering a frequency response of the pre-filter.

15 [27] A sampling rate converter as set forth in claim 24, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

20 [28] A sampling rate converter as set forth in claim 24, further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling
25 frequency of the output.

[29] A sampling rate converter comprising:

a convolution processing unit including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input
5 sample signals and a poly-phase filter having a selected coefficient,

a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter, and
10 a linear interpolation block for finding the value at the required position from linear interpolation, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the
15 impulse response becomes the filter coefficient, and a transmission function $H(z)$ is associated with a transmission function $Z(z)$ of a pre-filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired
20 characteristic in relation to a frequency response of the pre-filter.

[30] A sampling rate converter as set forth in claim 29, wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by
25 performing weighted approximation with respect to a desired characteristic in relation to a frequency

response of the pre-filter.

[31] A sampling rate converter as set forth in claim 29, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex

- 5 Exchange algorithm considering a frequency response of the pre-filter.

[32] A sampling rate converter as set forth in claim 29, further including a low bandpass filter preventing an aliasing component from occurring and folding from

- 10 occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

[33] A sampling rate converter as set forth in claim 29, further including a low bandpass filter preventing an imaging component from occurring and a non-original

- 15 frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

[34] A sampling rate converter as set forth in claim 29, wherein the selector includes a counter by which at least

- 20 a coefficient of linear interpolation, a number of a coefficient set of poly-phases, and a number of input samples are found.

[35] A sampling rate converter comprising:

- a convolution processing unit including poly-phase
25 filters able to set different filter coefficients
obtained by poly-phase decomposing a predetermined FIR

filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient,

a selector for selecting two points of samples
5 required for an output sample and selecting the coefficient of the corresponding poly-phase filter, and
a linear interpolation block for finding the value at the required position from linear interpolation,
wherein

10 the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and

the filter coefficient is set by performing weighted approximation with respect to a desired
15 characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[36] A sampling rate converter as set forth in claim 35,
wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by
20 performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[37] A sampling rate converter as set forth in claim 35,
further including a low bandpass filter preventing an
25 aliasing component from occurring and folding from occurring when the sampling frequency of the input is

lower than a sampling frequency of the output.

[38] A sampling rate converter as set forth in claim 35,
further including a low bandpass filter preventing an
imaging component from occurring and a non-original
5 frequency component from occurring when the sampling
frequency of the input is higher than a sampling
frequency of the output.

[39] A sampling rate converter as set forth in claim 35,
wherein the selector includes a counter by which at least
10 a coefficient of linear interpolation, a number of a
coefficient set of poly-phases, and a number of input
samples are found.

[40] A sampling rate converter comprising:

a convolution processing unit including poly-phase
15 filters able to set different filter coefficients
obtained by poly-phase decomposing a predetermined FIR
filter and performing convolution processing of input
sample signals and a poly-phase filter having a selected
coefficient,

20 a selector for selecting two points of samples
required for an output sample and selecting the
coefficient of the corresponding poly-phase filter, and
a linear interpolation block for finding the value
at the required position from linear interpolation,
25 wherein

the FIR filter is an FIR filter where an impulse

response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function $H(z)$ is associated with a transmission function $Z(z)$ of a pre-filter, and

5 the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

[41] A sampling rate converter as set forth in claim 40,
10 wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter.

15 [42] A sampling rate converter as set forth in claim 40, wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm passing any frequency point and considering a frequency response of the pre-filter.

20 [43] A sampling rate converter as set forth in claim 40, further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output.

25 [44] A sampling rate converter as set forth in claim 40, further including a low bandpass filter preventing an

imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output.

- 5 [45] A sampling rate converter as set forth in claim 40, wherein the selector includes a counter by which at least a coefficient of linear interpolation, a number of a coefficient set of poly-phases, and a number of input samples are found.
- 10 [46] A sampling rate conversion method comprising:
a first step of inserting $U-1$ zero points between sample signals and raising the sampling frequency U -fold,
a second step of performing predetermined convolution processing with respect to a signal
15 multiplied in its sampling frequency by U by a convolution processing unit including an FIR filter in which an impulse response is expressed by a finite time length, an impulse response becomes the filter coefficient, and a transmission function $H(z)$ is
20 associated with a transmission function $Z(z)$ of a pre-filter, and
a third step of selecting two points of samples with respect to the results of processing and finding the value at the required position from linear interpolation,
25 wherein
the filter coefficient of the FIR filter is

calculated by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[47] A sampling rate conversion method comprising:

5 a first step of inserting $U-1$ zero points between sample signals and raising the sampling frequency U -fold, a second step of performing predetermined convolution processing with respect to a signal multiplied in its sampling frequency by U by a
10 convolution processing unit including an FIR filter in which an impulse response is expressed by a finite time length and an impulse response becomes the filter coefficient, and

 a third step of selecting two points of samples
15 with respect to the results of processing and finding the value at the required position from linear interpolation, wherein

 the filter coefficient of the FIR filter is calculated by performing weighted approximation with
20 respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point.

[48] A sampling rate conversion method comprising:

 a first step of inserting $U-1$ zero points between
25 sample signals and raising the sampling frequency U -fold, a second step of performing predetermined

convolution processing with respect to a signal
multiplied in its sampling frequency by U by a
convolution processing unit including an FIR filter in
which an impulse response is expressed by a finite time
5 length, an impulse response becomes the filter
coefficient, and a transmission function $H(z)$ is
associated with a transmission function $Z(z)$ of a pre-
filter, and

a third step of selecting two points of samples
10 with respect to the results of processing and finding the
value at the required position from linear interpolation,
wherein

the filter coefficient of the FIR filter is
calculated by performing weighted approximation with
15 respect to a desired characteristic in relation to
frequency points to be passed and a frequency response of
the pre-filter.

[49] A sampling rate conversion method comprising:

a first step of performing convolution processing
20 of input sample signals and poly-phase filters decomposed
to poly-phases by a plurality of convolution processing
units including poly-phase filters obtained by poly-phase
decomposing a predetermined FIR filter,

a second step of inserting $U-1$ zero points between
25 output signals of corresponding convolution processing
units and raising the sampling frequency U -fold,

a third step of adjusting the propagation time of a plurality of signals having sampling frequencies raised U -fold and generating a signal obtained by adding all signals, and

5 a fourth step of selecting two points of samples with respect to the signal by the third step and finding the value at the required position from the linear interpolation, wherein

the FIR filter is a FIR filter where an impulse
10 response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function $H(z)$ is associated with a transmission function $Z(z)$ of a pre-filter, and

the filter coefficient is calculated by performing
15 weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[50] A sampling rate conversion method comprising:

a first step of performing convolution processing
20 of input sample signals and poly-phase filters decomposed to poly-phases by a plurality of convolution processing units including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter,

a second step of inserting $U-1$ zero points between
25 output signals of corresponding convolution processing units and raising the sampling frequency U -fold,

a third step of adjusting the propagation time of a plurality of signals having sampling frequencies raised U-fold and generating a signal obtained by adding all signals, and

5 a fourth step of selecting two points of samples with respect to the signal by the third step and finding the value at the required position from the linear interpolation, wherein

the FIR filter is a FIR filter where an impulse
10 response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and

the filter coefficient is calculated by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive
15 condition so as to pass any frequency point.

[51] A sampling rate conversion method comprising:

a first step of performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases by a plurality of convolution processing
20 units including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter,

a second step of inserting U-1 zero points between output signals of corresponding convolution processing units and raising the sampling frequency U-fold,

25 a third step of adjusting the propagation time of a plurality of signals having sampling frequencies raised

U-fold and generating a signal obtained by adding all signals, and

a fourth step of selecting two points of samples with respect to the signal by the third step and finding
5 the value at the required position from the linear interpolation, wherein

the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a
10 transmission function $H(z)$ is associated with a transmission function $Z(z)$ of a pre-filter, and

the filter coefficient is calculated by performing weighted approximation with respect to a desired characteristic in relation to a frequency points to be
15 passed and a frequency response of the pre-filter.

[52] A sampling rate conversion method comprising:

a first step of selecting two points of samples required for an output sample and selecting a coefficient of a corresponding poly-phase filter and
20 a second step of performing convolution processing of input sample signals and the poly-phase filter having the selected coefficient by a convolution processing unit including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter and able to set
25 different filter coefficients, wherein

the FIR filter is a FIR filter where an impulse

response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function $H(z)$ is associated with a transmission function $Z(z)$ of a pre-filter, and

5 the filter coefficient is calculated by performing the weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter.

[53] A sampling rate conversion method comprising:

10 a first step of selecting two points of samples required for an output sample and selecting a coefficient of a corresponding poly-phase filter and

 a second step of performing convolution processing of input sample signals and the poly-phase filter having
15 the selected coefficient by a convolution processing unit including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter and able to set different filter coefficients, wherein

 the FIR filter is a FIR filter where an impulse
20 response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and

 the filter coefficient is calculated by performing the weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive
25 condition so as to pass any frequency point.

[54] A sampling rate conversion method comprising:

a first step of selecting two points of samples required for an output sample and selecting a coefficient of a corresponding poly-phase filter and

a second step of performing convolution processing
5 of input sample signals and the poly-phase filter having the selected coefficient by a convolution processing unit including poly-phase filters obtained by poly-phase decomposing a predetermined FIR filter and able to set different filter coefficients, wherein

10 the FIR filter is a FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function $H(z)$ is associated with a transmission function $Z(z)$ of a pre-filter, and

15 the filter coefficient is calculated by performing the weighted approximation with respect to a desired characteristic in relation to a frequency points to be passed and a frequency response of the pre-filter.

[55] An audio apparatus including a sampling rate
20 converter, wherein the sampling rate converter comprises an up sampler for inserting $U-1$ zero points between sample signals and raising a sampling frequency U -fold, a convolution processing unit including an FIR filter and performing predetermined convolution
25 processing with respect to an output signal of the up sampler, and

a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from linear interpolation,
5 wherein

the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function $H(z)$ is
10 associated with a transmission function $Z(z)$ of a pre-filter, and

the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency to be passed
15 and/or a frequency response of the pre-filter.

[56] An audio apparatus including a sampling rate converter, wherein the sampling rate converter comprises

a plurality of convolution processing units including pre-phase filters obtained by poly-phase
20 decomposing a predetermined FIR filter and performing convolution processing of input sample signals and poly-phase filters decomposed to poly-phases,

a plurality of up samplers for inserting $U-1$ zero points between output signals of corresponding
25 convolution processing units and raising the sampling frequency U -fold,

an adding means for generating a signal after adding all signals by adjusting a propagation time of output signals of the plurality of up samplers, and

a linear interpolation block for selecting two
5 points of samples with respect to the signal by the adding means and finding the value at the required position from linear interpolation, wherein

the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, an impulse
10 response becomes the filter coefficient, and a transmission function $H(z)$ is associated with a transmission function $Z(z)$ of a pre-filter, and

the filter coefficient is set by performing the weighted approximation with respect to a desired
15 characteristic in relation to a frequency to be passed and/or a frequency response of the pre-filter.

[57] An audio apparatus including a sampling rate converter, wherein the sampling rate converter comprises

a convolution processing unit including poly-phase
20 filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient and

25 a selector for selecting two points of samples required for an output sample and selecting the

coefficient of the corresponding poly-phase filter,
wherein

the FIR filter is an FIR filter where an impulse
response is expressed by a finite time length, the
5 impulse response becomes the filter coefficient, and a
transmission function $H(z)$ is associated with a
transmission function $Z(z)$ of a pre-filter, and

the filter coefficient is set by performing
weighted approximation with respect to a desired
10 characteristic in relation to a frequency to be passed
and/or a frequency response of the pre-filter.